Anti-Aliasing, Analog Filters for Data Acquisition Systems

Author: Bonnie C. Baker
Microchip Technology Inc.

INTRODUCTION

Analog filters can be found in almost every electronic circuit. Audio systems use them for preamplification, equalization, and tone control. In communication systems, filters are used for tuning in specific frequencies and eliminating others. Digital signal processing systems use filters to prevent the aliasing of out-of-band noise and interference.

This application note investigates the design of analog filters that reduce the influence of extraneous noise in data acquisition systems. These types of systems primarily utilize low-pass filters, digital filters or a combination of both. With the analog low-pass filter, high frequency noise and interference can be removed from the signal path prior to the analog-to-digital (A/D) conversion. In this manner, the digital output code of the conversion does not contain undesirable aliased harmonic information. In contrast, a digital filter can be utilized to reduce in-band frequency noise by using averaging techniques.

Although the application note is about analog filters, the first section will compare the merits of an analog filtering strategy versus digital filtering.

Following this comparison, analog filter design parameters are defined. The frequency characteristics of a low pass filter will also be discussed with some reference to specific filter designs. In the third section, low pass filter designs will be discussed in depth.

The next portion of this application note will discuss techniques on how to determine the appropriate filter design parameters of an anti-aliasing filter. In this section, aliasing theory will be discussed. This will be followed by operational amplifier filter circuits. Examples of active and passive low pass filters will also be discussed. Finally, a 12-bit circuit design example will be given. All of the active analog filters discussed in this application note can be designed using Microchip's FilterLab software. FilterLab will calculate capacitor and resistor values, as well as, determine the number of poles that are required for the application. The program will also generate a SPICE macromodel, which can be used for spice simulations.

ANALOG VERSUS DIGITAL FILTERS

A system that includes an analog filter, a digital filter or both is shown in Figure 1. When an analog filter is implemented, it is done prior to the analog-to-digital conversion. In contrast, when a digital filter is implemented, it is done after the conversion from analog-to-digital has occurred. It is obvious why the two filters are implemented at these particular points, however, the ramifications of these restrictions are not quite so obvious.

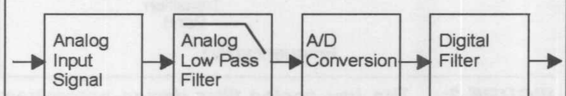


FIGURE 1: The data acquisition system signal chain can utilize analog or digital filtering techniques or a combination of the two.

There are a number of system differences when the filtering function is provided in the digital domain rather than the analog domain and the user should be aware of these.

Analog filtering can remove noise superimposed on the analog signal before it reaches the Analog-to-Digital Converter. In particular, this includes extraneous noise peaks. Digital filtering cannot eliminate these peaks riding on the analog signal. Consequently, noise peaks riding on signals near full scale have the potential to saturate the analog modulator of the A/D Converter. This is true even when the average value of the signal is within limits.

Additionally, analog filtering is more suitable for higher speed systems, i.e., above approximately 5kHz. In these types of systems, an analog filter can reduce noise in the out-of-band frequency region. This, in turn, reduces fold back signals (see the "Anti-Aliasing Filter Theory" section in this application note). The task of obtaining high resolution is placed on the A/D Converter. In contrast, a digital filter, by definition uses oversampling and averaging techniques to reduce in band and out of band noise. These two processes take time.

Since digital filtering occurs after the A/D conversion process, it can remove noise injected during the conversion process. Analog filtering cannot do this. Also, the digital filter can be made programmable far more

© 1999 Microchip Technology Inc.

readily than an analog filter. Depending on the digital filter design, this gives the user the capability of programming the cutoff frequency and output data rates.

KEY LOW PASS ANALOG FILTER DESIGN PARAMETERS

A low pass analog filter can be specified with four parameters as shown in Figure 2 ($f_{CUT-OFF}$, f_{STOP}, A_{MAX}, and M).

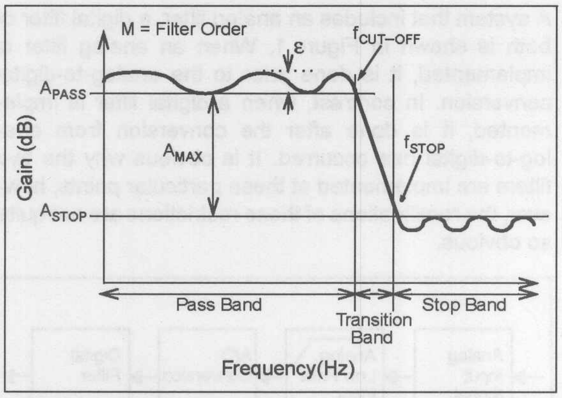


FIGURE 2: *The key analog filter design parameters include the –3dB cut-off frequency of the filter ($f_{cut-off}$), the frequency at which a minimum gain is acceptable (f_{stop}) and the number of poles (M) implemented with the filter.*

The cut-off frequency ($f_{CUT-OFF}$) of a low pass filter is defined as the -3dB point for a Butterworth and Bessel filter or the frequency at which the filter response leaves the error band for the Chebyshev.

The frequency span from DC to the cut-off frequency is defined as the pass band region. The magnitude of the response in the pass band is defined as A_{PASS} as shown in Figure 2. The response in the pass band can be flat with no ripple as is when a Butterworth or Bessel filter is designed. Conversely, a Chebyshev filter has a ripple up to the cut-off frequency. The magnitude of the ripple error of a filter is defined as ε.

By definition, a low pass filter passes lower frequencies up to the cut-off frequency and attenuates the higher frequencies that are above the cut-off frequency. An important parameter is the filter system gain, A_{MAX}. This is defined as the difference between the gain in the pass band region and the gain that is achieved in the stop band region or $A_{MAX} = A_{PASS} - A_{STOP}$

In the case where a filter has ripple in the pass band, the gain of the pass band (A_{PASS}) is defined as the bottom of the ripple. The stop band frequency, f_{STOP} , is the frequency at which a minimum attenuation is reached. Although it is possible that the stop band has a ripple, the minimum gain (A_{STOP}) of this ripple is defined at the highest peak.

As the response of the filter goes beyond the cut-off frequency, it falls through the transition band to the stop band region. The bandwidth of the transition band is determined by the filter design (Butterworth, Bessel, Chebyshev, etc.) and the order (M) of the filter. The filter order is determined by the number of poles in the transfer function. For instance, if a filter has three poles in its transfer function, it can be described as a 3rd order filter.

Generally, the transition bandwidth will become smaller when more poles are used to implement the filter design. This is illustrated with a Butterworth filter in Figure 3. Ideally, a low-pass, anti-aliasing filter should perform with a "brick wall" style of response, where the transition band is designed to be as small as possible. Practically speaking, this may not be the best approach for an anti-aliasing solution. With active filter design, every two poles require an operational amplifier. For instance, if a 32nd order filter is designed, 16 operational amplifiers, 32 capacitors and up to 64 resistors would be required to implement the circuit. Additionally, each amplifier would contribute offset and noise errors into the pass band region of the response.

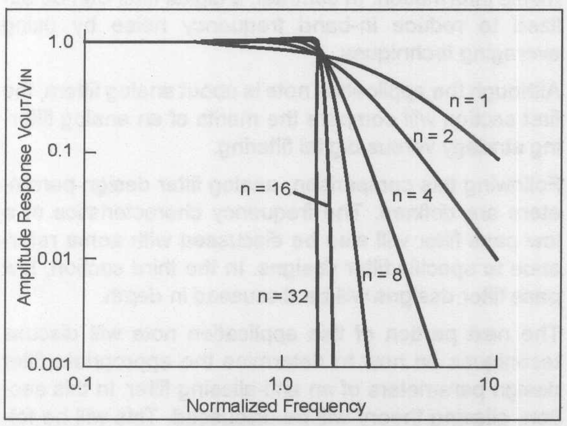


FIGURE 3: *A Butterworth design is used in a low pass filter implementation to obtain various responses with frequency dependent on the number of poles or order (M) of the filter.*

Strategies on how to work around these limitations will be discussed in the "Anti-Aliasing Theory" section of this application note.

© 1999 Microchip Technology Inc.

ANALOG FILTER DESIGNS

The more popular filter designs are the Butterworth, Bessel, and Chebyshev. Each filter design can be identified by the four parameters illustrated in Figure 2. Other filter types not discussed in this application note include Inverse Chebyshev, Elliptic, and Cauer designs.

Butterworth Filter

The Butterworth filter is by far the most popular design used in circuits. The transfer function of a Butterworth filter consists of all poles and no zeros and is equated to:

$$V_{OUT}/V_{IN} = G/(a_0 s^n + a_1 s^{n-1} + a_2 s^{n-2} \ldots a_{n-1} s^2 + a_n s + 1)$$

where G is equal to the gain of the system.

Table 1 lists the denominator coefficients for a Butterworth design. Although the order of a Butterworth filter design theoretically can be infinite, this table only lists coefficients up to a 5th order filter.

M	a_0	a_1	a_2	a_3	a_4
2	1.0	1.4142136			
3	1.0	2.0	2.0		
4	1.0	2.6131259	3.4142136	2.6131259	
5	1.0	3.2360680	5.2360680	5.2360680	3.2360680

TABLE 1: *Coefficients versus filter order for Butterworth designs.*

As shown in Figure 4a., the frequency behavior has a maximally flat magnitude response in pass-band. The rate of attenuation in transition band is better than Bessel, but not as good as the Chebyshev filter. There is no ringing in stop band. The step response of the Butterworth is illustrated in Figure 5a. This filter type has some overshoot and ringing in the time domain, but less than the Chebyshev.

Chebyshev Filter

The transfer function of the Chebyshev filter is only similar to the Butterworth filter in that it has all poles and no zeros with a transfer function of:

$$V_{OUT}/V_{IN} = G/(a_0 + a_1 s + a_2 s^2 + \ldots a_{n-1} s^{n-1} + s_n)$$

Its frequency behavior has a ripple (Figure 4b.) in the pass-band that is determined by the specific placement of the poles in the circuit design. The magnitude of the ripple is defined in Figure 2 as ε. In general, an increase in ripple magnitude will lessen the width of the transition band.

The denominator coefficients of a 0.5dB ripple Chebyshev design are given in Table 2. Although the order of a Chebyshev filter design theoretically can be infinite, this table only lists coefficients up to a 5th order filter.

M	a_0	a_1	a_2	a_3	a_4
2	1.516203	1.425625			
3	0.715694	1.534895	1.252913		
4	0.379051	1.025455	1.716866	1.197386	
5	0.178923	0.752518	1.309575	1.937367	1.172491

TABLE 2: *Coefficients versus filter order for 1/2dB ripple Chebyshev designs.*

The rate of attenuation in the transition band is steeper than Butterworth and Bessel filters. For instance, a 5th order Butterworth response is required if it is to meet the transition band width of a 3rd order Chebyshev. Although there is ringing in the pass band region with this filter, the stop band is void of ringing. The step response (Figure 5b.) has a fair degree of overshoot and ringing.

Bessel Filter

Once again, the transfer function of the Bessel filter has only poles and no zeros. Where the Butterworth design is optimized for a maximally flat pass band response and the Chebyshev can be easily adjusted to minimize the transition bandwidth, the Bessel filter produces a constant time delay with respect to frequency over a large range of frequency. Mathematically, this relationship can be expressed as:

$$C = -\Delta\theta * \Delta f$$

where:

C is a constant,

θ is the phase in degrees, and

f is frequency in Hz

Alternatively, the relationship can be expressed in degrees per radian as:

$$C = -\Delta\theta / \Delta\omega$$

where:

C is a constant,

θ is the phase in degrees, and

ω is in radians.

The transfer function for the Bessel filter is:

$$V_{OUT}/V_{IN} = G/(a_0 + a_1 s + a_2 s^2 + \ldots a_{n-1} s^{n-1} + s_n)$$

The denominator coefficients for a Bessel filter are given in Table 3. Although the order of a Bessel filter design theoretically can be infinite, this table only lists coefficients up to a 5th order filter.

M	a_0	a_1	a_2	a_3	a_4
2	3	3			
3	15	15	6		
4	105	105	45	10	
5	945	945	420	105	15

TABLE 3: *Coefficients versus filter order for Bessel designs.*

The Bessel filter has a flat magnitude response in pass-band (Figure 4c). Following the pass band, the rate of attenuation in transition band is slower than the Butterworth or Chebyshev. And finally, there is no ringing in stop band. This filter has the best step response of all the filters mentioned above, with very little overshoot or ringing (Figure 5c.).

© 1999 Microchip Technology Inc.

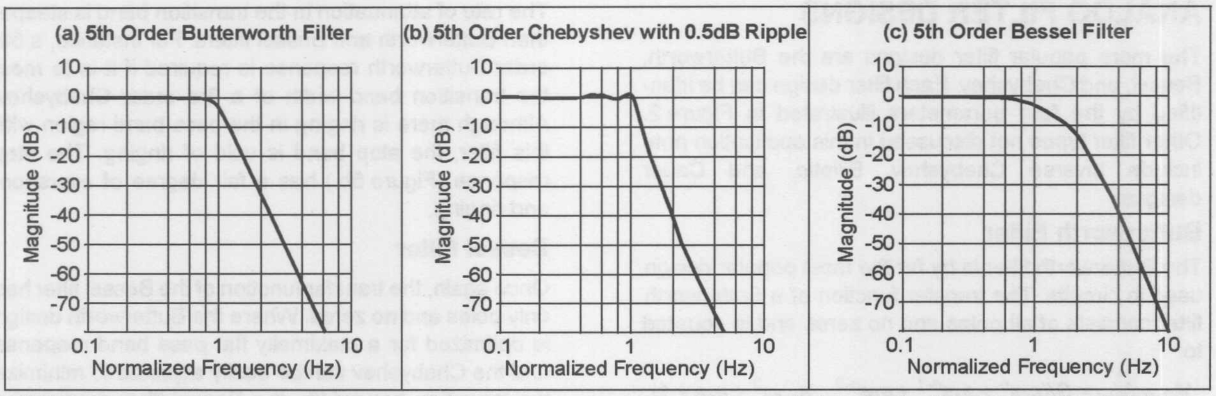


FIGURE 4: *The frequency responses of the more popular filters, Butterworth (a), Chebyshev (b), and Bessel (c)..*

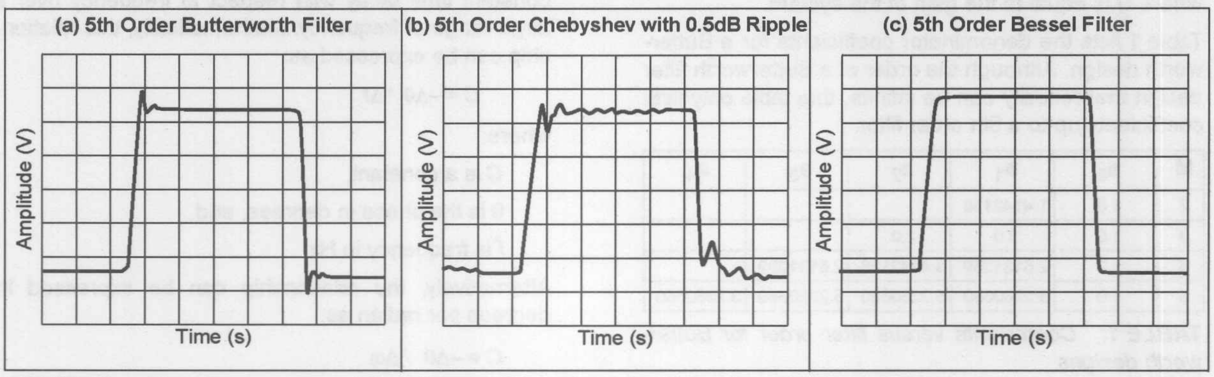


FIGURE 5: *The step response of the 5th order filters shown in Figure 4 are illustrated here.*

ANTI-ALIASING FILTER THEORY

A/D Converters are usually operated with a constant sampling frequency when digitizing analog signals. By using a sampling frequency (f_S), typically called the Nyquist rate, all input signals with frequencies below $f_S/2$ are reliably digitized. If there is a portion of the input signal that resides in the frequency domain above $f_S/2$, that portion will fold back into the bandwidth of interest with the amplitude preserved. The phenomena makes it impossible to discern the difference between a signal from the lower frequencies (below $f_S/2$) and higher frequencies (above $f_S/2$).

This aliasing or fold back phenomena is illustrated in the frequency domain in Figure 6.

In both parts of this figure, the x-axis identifies the frequency of the sampling system, f_S. In the left portion of Figure 6, five segments of the frequency band are identified. Segment N =0 spans from DC to one half of the sampling rate. In this bandwidth, the sampling system will reliably record the frequency content of an analog input signal. In the segments where N > 0, the frequency content of the analog signal will be recorded by the digitizing system in the bandwidth of the segment N = 0. Mathematically, these higher frequencies will be folded back with the following equation:

$$f_{ALIASED} = |f_{IN} - Nf_S|$$

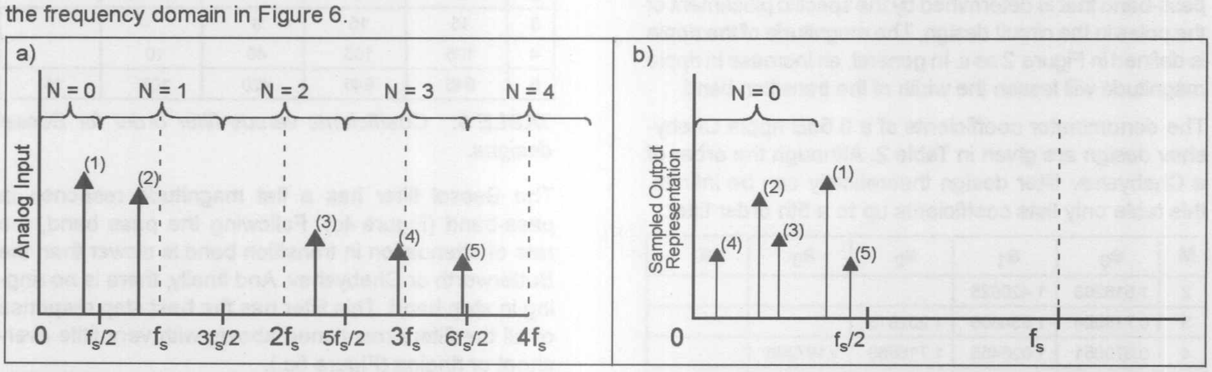


FIGURE 6: *A system that is sampling an input signal at f_s (a) will identify signals with frequencies below $f_s/2$ as well as above. Input signals below $f_s/2$ will be reliably digitized while signals above $f_s/2$ will be folded back (b) and appear as lower frequencies in the digital output.*

© 1999 Microchip Technology Inc.

For example, let the sampling rate, (f_S), of the system be equal to 100kHz and the frequency content of:

$f_{IN}(1) = 41kHz$

$f_{IN}(2) = 82kHz$

$f_{IN}(3) = 219kHz$

$f_{IN}(4) = 294kHz$

$f_{IN}(5) = 347kHz$

The sampled output will contain accurate amplitude information of all of these input signals, however, four of them will be folded back into the frequency range of DC to $f_S/2$ or DC to 50kHz. By using the equation $f_{OUT} = |f_{IN} - Nf_S|$, the frequencies of the input signals are transformed to:

$f_{OUT}(1) = |41kHz - 0 \times 100kHz| = 41kHz$

$f_{OUT}(2) = |82kHz - 1 \times 100kHz| = 18kHz$

$f_{OUT}(3) = |219kHz - 2 \times 100kHz| = 19kHz$

$f_{OUT}(4) = |294kHz - 3 \times 100kHz| = 6kHz$

$f_{OUT}(5) = |347kHz - 4 \times 100kHz| = 53kHz$

Note that all of these signal frequencies are between DC and $f_S/2$ and that the amplitude information has been reliably retained.

This frequency folding phenomena can be eliminated or significantly reduced by using an analog low pass filter prior to the A/D Converter input. This concept is illustrated in Figure 7. In this diagram, the low pass filter attenuates the second portion of the input signal at frequency (2). Consequently, this signal will not be aliased into the final sampled output. There are two regions of the analog low pass filter illustrated in Figure 7. The region to the left is within the bandwidth of DC to $f_S/2$. The second region, which is shaded, illustrates the transition band of the filter. Since this region is greater than $f_S/2$, signals within this frequency band will be aliased into the output of the sampling system. The affects of this error can be minimized by moving the corner frequency of the filter lower than $f_S/2$ or increasing the order of the filter. In both cases, the minimum gain of the filter, A_{STOP}, at $f_S/2$ should less than the signal-to-noise ratio (SNR) of the sampling system.

For instance, if a 12-bit A/D Converter is used, the ideal SNR is 74dB. The filter should be designed so that its gain at f_{STOP} is at least 74dB less than the pass band gain. Assuming a 5th order filter is used in this example:

$f_{CUT\text{-}OFF} = 0.18 f_S/2$ *for a Butterworth Filter*

$f_{CUT\text{-}OFF} = 0.11 f_S/2$ *for a Bessel Filter*

$f_{CUT\text{-}OFF} = 0.21 f_S/2$ *for a Chebyshev Filter with 0.5dB ripple in the pass band*

$f_{CUT\text{-}OFF} = 0.26 f_S/2$ *for a Chebyshev Filter with 1dB ripple in the pass band*

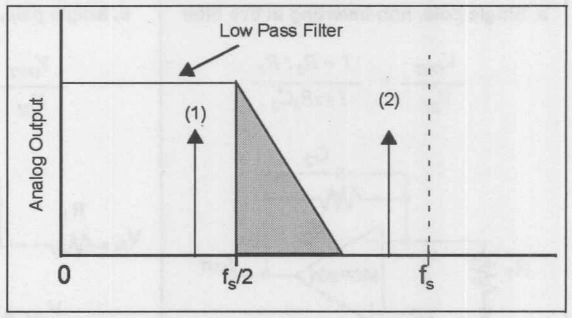


FIGURE 7: *If the sampling system has a low pass analog filter prior to the sampling mechanism, high frequency signals will be attenuated and not sampled.*

ANALOG FILTER REALIZATION

Traditionally, low pass filters were implemented with passive devices, ie. resistors and capacitors. Inductors were added when high pass or band pass filters were needed. At the time active filter designs were realizable, however, the cost of operational amplifiers was prohibitive. Passive filters are still used with filter design when a single pole filter is required or where the bandwidth of the filter operates at higher frequencies than leading edge operational amplifiers. Even with these two exceptions, filter realization is predominately implemented with operational amplifiers, capacitors and resistors.

Passive Filters

Passive, low pass filters are realized with resistors and capacitors. The realization of single and double pole low pass filters are shown in Figure 8.

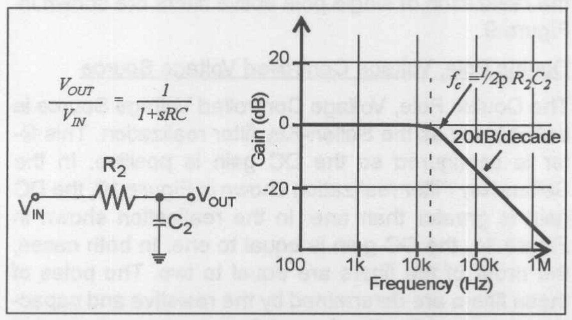


FIGURE 8: *A resistor and capacitor can be used to implement a passive, low pass analog filter. The input and output impedance of this type of filter implementation is equal to R_2.*

The output impedance of a passive low pass filter is relatively high when compared to the active filter realization. For instance, a 1kHz low pass filter which uses a 0.1µF capacitor in the design would require a 1.59kΩ resistor to complete the implementation. This value of resistor could create an undesirable voltage drop or make impedance matching difficult. Consequently, passive filters are typically used to implement a single pole. Single pole operational amplifier filters have the added benefit of "isolating" the high impedance of the filter from the following circuitry.

© 1999 Microchip Technology Inc.

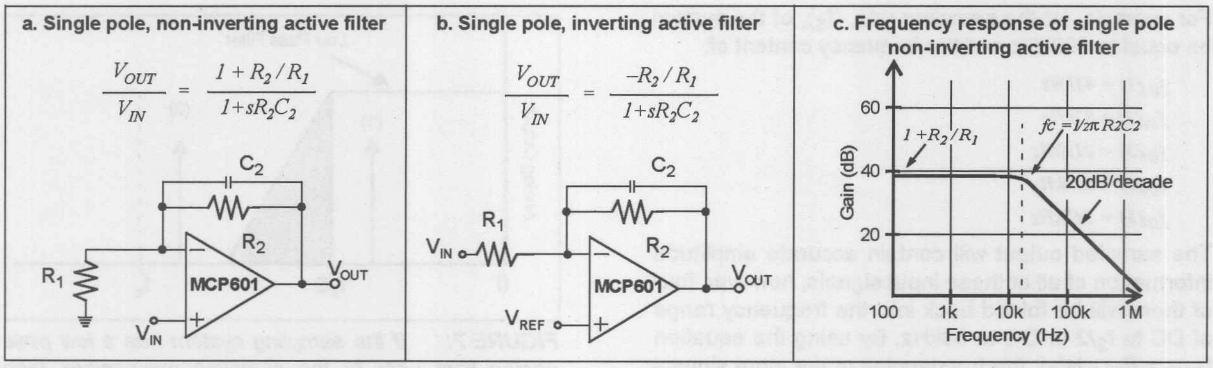


FIGURE 9: *An operational amplifier in combination with two resistors and one capacitor can be used to implement a 1st order filter. The frequency response of these active filters is equivalent to a single pole passive low pass filter.*

It is very common to use a single pole, low pass, passive filter at the input of a Delta-Sigma A/D Converter. In this case, the high output impedance of the filter does not interfere with the conversion process.

Active Filters

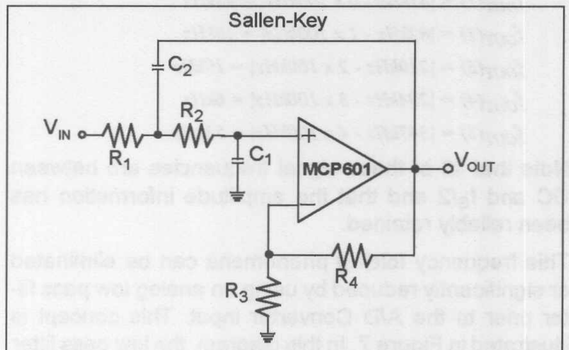
An active filter uses a combination of one amplifier, one to three resistors and one to two capacitors to implement one or two poles. The active filter offers the advantage of providing "isolation" between stages. This is possible by taking advantage of the high input impedance and low output impedance of the operational amplifier. In all cases, the order of the filter is determined by the number of capacitors at the input and in the feedback loop of the amplifier.

Single Pole Filter

The frequency response of the single pole, active filter is identical to a single pole passive filter. Examples of the realization of single pole active filters are shown in Figure 9.

Double Pole, Voltage Controlled Voltage Source

The Double Pole, Voltage Controlled Voltage Source is better know as the Sallen-Key filter realization. This filter is configured so the DC gain is positive. In the Sallen-Key Filter realization shown in Figure 10, the DC gain is greater than one. In the realization shown in Figure 11, the DC gain is equal to one. In both cases, the order of the filters are equal to two. The poles of these filters are determined by the resistive and capacitive values of R_1, R_2, C_1 and C_2.



$$\frac{V_{OUT}}{V_{IN}} = \frac{K/(R_1R_2C_1C_2)}{s^2+s(1/R_1C_2+1/R_2C_2+1/R_2C_1-K/R_2C_1+1/R_1R_2C_1C_2)}$$

$$K = 1 + R_4/R_3$$

FIGURE 10: *The double pole or Sallen-Key filter implementation has a gain $G = 1 + R_4/R_3$. If R_3 is open and R_4 is shorted the DC gain is equal to 1 V/V.*

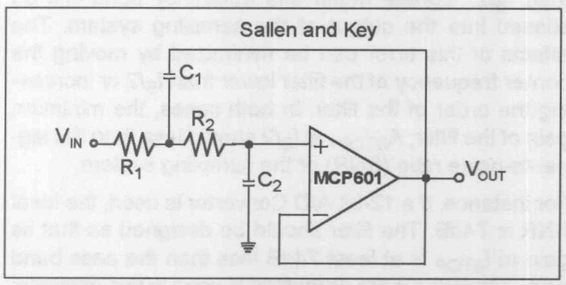
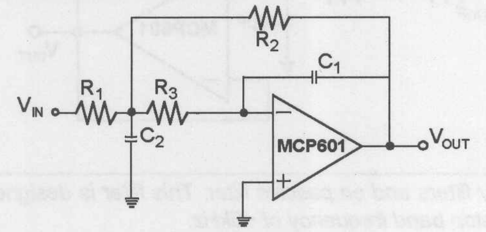


FIGURE 11: *The double pole or Sallen-Key filter implementation with a DC gain is equal to 1V/V.*

© 1999 Microchip Technology Inc.

Double Pole Multiple Feedback

The double pole, multiple feedback realization of a 2nd order low pass filter is shown in Figure 12. This filter can also be identified as simply a Multiple Feedback Filter. The DC gain of this filter inverts the signal and is equal to the ratio of R_1 and R_2. The poles are determined by the values of R_1, R_3, C_1, and C_2.



$$\frac{V_{OUT}}{V_{IN}} = \frac{-1/R_1R_3C_5C_6}{s^2C_2C_1 + sC_1(1/R_1 + 1/R_2 + 1/R_3) + 1/(R_2R_3C_2C_1)}$$

FIGURE 12: *A double pole, multiple feedback circuit implementation uses three resistors and two capacitors to implement a 2nd order analog filter. DC gain is equal to* $-R_2 / R_1$.

ANTI-ALIASING FILTER DESIGN EXAMPLE

In the following examples, the data acquisition system signal chain shown in Figure 1 will be modified as follows. The analog signal will go directly into an active low pass filter. In this example, the bandwidth of interest of the analog signal is DC to 1kHz. The low pass filter will be designed so that high frequency signals from the analog input do not pass through to the A/D Converter in an attempt to eliminate aliasing errors. The implementation and order of this filter will be modified according to the design parameters. Excluding the filtering function, the anti-aliasing filter will not modify the signal further, i.e., implement a gain or invert the signal. The low pass filter segment will be followed by a 12-bit SAR A/D Converter. The sampling rate of the A/D Converter will be 20kHz, making 1/2 of Nyquist equal to 10kHz. The ideal signal-to-noise ratio of a 12-bit A/D Converter of 74dB. This design parameter will be used when determining the order of the anti-aliasing filter. The filter examples discussed in this section were generated using Microchip's FilterLab software.

Three design parameters will be used to implement appropriate anti-aliasing filters:

1. Cut-off frequency for filter must be 1kHz or higher.
2. Filter attenuates the signal to -74dB at 10kHz.
3. The analog signal will only be filtered and not gained or inverted.

Implementation with Bessel Filter Design

A Bessel Filter design is used in Figure 13 to implement the anti-aliasing filter in the system described above. A 5th order filter that has a cut-off frequency of 1kHz is required for this implementation. A combination of two Sallen-Key filters plus a passive low pass filter are designed into the circuit as shown in Figure 14. This filter attenuates the analog input signal 79dB from the pass band region to 10kHz. The frequency response of this Bessel, 5th order filter is shown in Figure 13.

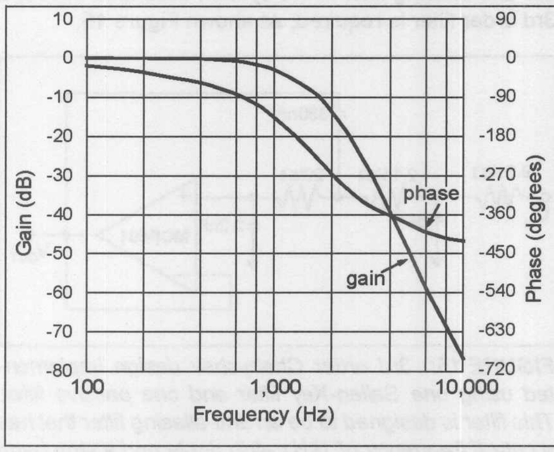


FIGURE 13: *Frequency response of 5th order Bessel design implemented in Figure 14.*

© 1999 Microchip Technology Inc.

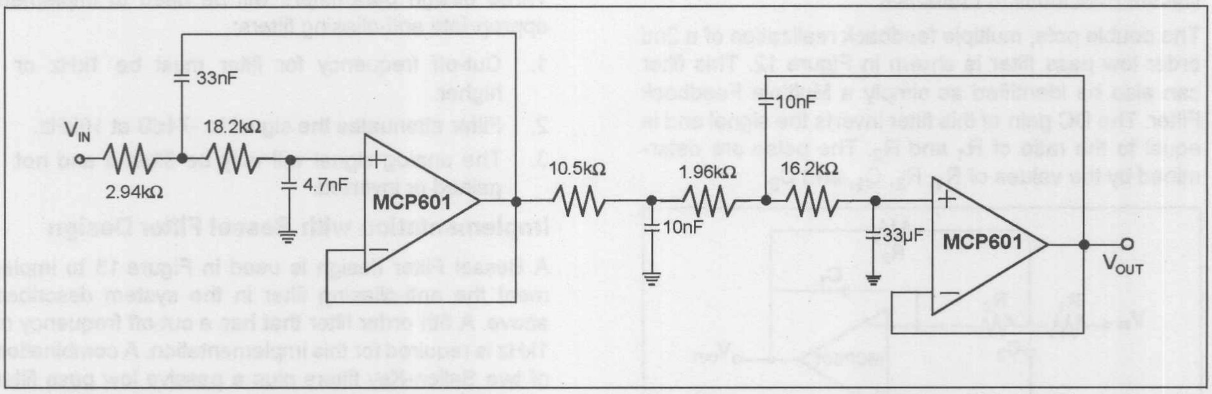


FIGURE 14: 5th order Bessel design implemented two Sallen-Key filters and on passive filter. This filter is designed to be an anti-aliasing filter that has a cut-off frequency of 1kHz and a stop band frequency of ~5kHz.

Implementation with Chebyshev Design

When a Chebyshev filter design is used to implement the anti-aliasing filter in the system described above, a 3rd order filter is required, as shown Figure 15.

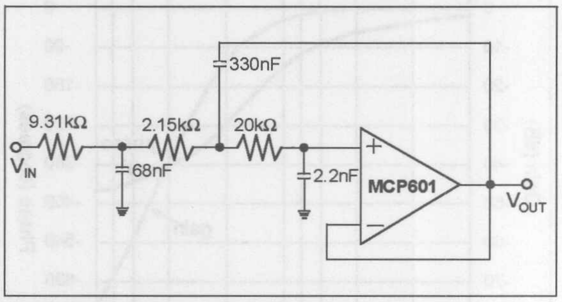


FIGURE 15: 3rd order Chebyshev design implemented using one Sallen-Key filter and one passive filter. This filter is designed to be an anti-aliasing filter that has a cut-off frequency of 1kHz -4db ripple and a stop band frequency of ~5kHz.

Although the order of this filter is less than the Bessel, it has a 4dB ripple in the pass band portion of the frequency response. The combination of one Sallen-Key filter plus a passive low pass filter is used. This filter is attenuated to -70dB at 10kHz. The frequency response of this Chebyshev 3rd order filter is shown in Figure 16.

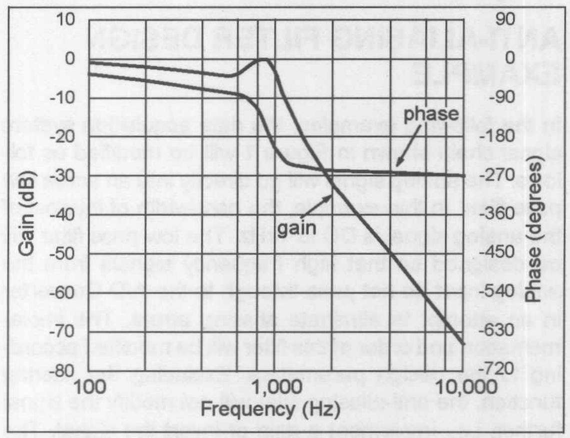


FIGURE 16: Frequency response of 3rd order Chebyshev design implemented in Figure 15.

This filter provides less than the ideal 74dB of dynamic range (A_{MAX}), which should be taken into consideration.

The difference between -70dB and -74dB attenuation in a 12-bit system will introduce little less than 1/2 LSB error. This occurs as a result of aliased signals from 10kHz to 11.8KHz. Additionally, a 4dB gain error will occur in the pass band. This is a consequence of the ripple response in the pass band, as shown in Figure 16.

© 1999 Microchip Technology Inc.

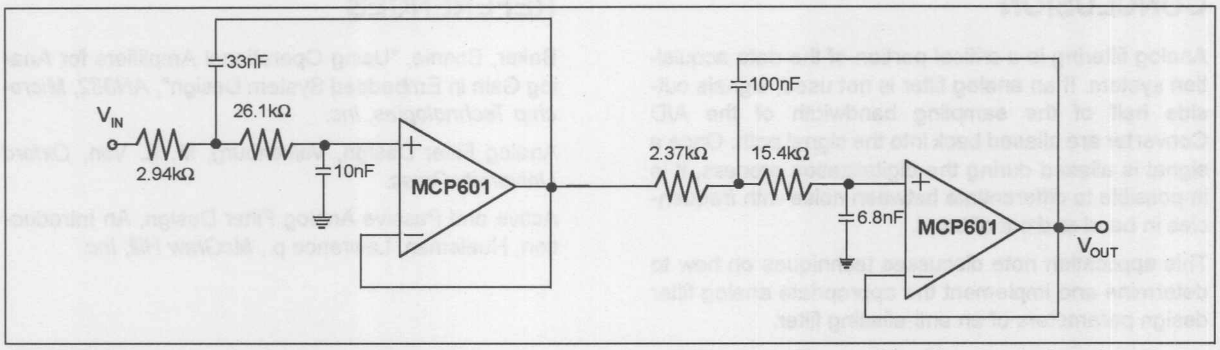


FIGURE 17: 4th order Butterworth design implemented two Sallen-Key filters. This filter is designed to be an anti-aliasing filter that has a cut-off frequency of 1kHz and a stop band frequency of ~5kHz.

Implementation with Butterworth Design

As a final alternative, a Butterworth filter design can be used in the filter implementation of the anti-aliasing filter, as shown in Figure 17.

For this circuit implementation, a 4th order filter is used with a cut-off frequency of 1kHz. Two Sallen-Key filters are used. This filter attenuates the pass band signal 80dB at 10kHz. The frequency response of this Butterworth 4th order filter is shown in Figure 18.

The frequency response of the three filters described above along with several other options are summarized in Table 4.

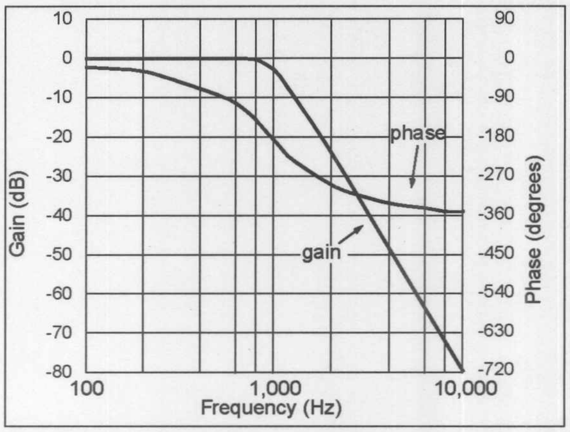


FIGURE 18: Frequency response of 4th order Butterworth design implemented in Figure 17.

FILTER ORDER, M	BUTTERWORTH, A_{MAX} (dB)	BESSEL, A_{MAX} (dB)	CHEBYSHEV, A_{MAX} (dB) W/ RIPPLE ERROR OF 1dB	CHEBYSHEV, A_{MAX} (dB) W/ RIPPLE ERROR OF 4dB
3	60	51	65	70
4	80	66	90	92
5	100	79	117	122
6	120	92	142	144
7	140	104	169	174

TABLE 4: Theoretical frequency response at 10kHz of various filter designs versus filter order. Each filter has a cut-off frequency of 1kHz.

CONCLUSION

Analog filtering is a critical portion of the data acquisition system. If an analog filter is not used, signals outside half of the sampling bandwidth of the A/D Converter are aliased back into the signal path. Once a signal is aliased during the digitalization process, it is impossible to differentiate between noise with frequencies in band and out of band.

This application note discusses techniques on how to determine and implement the appropriate analog filter design parameters of an anti-aliasing filter.

REFERENCES

Baker, Bonnie, "Using Operational Amplifiers for Analog Gain in Embedded System Design", *AN682, Microchip Technologies, Inc.*

Analog Filter Design, Valkenburg, M. E. Van, *Oxford University Press.*

Active and Passive Analog Filter Design, An Introduction, Huelsman, Lawrence p., *McGraw Hill, Inc.*

© 1999 Microchip Technology Inc.

NOTES:

Note the following details of the code protection feature on PICmicro® MCUs.

- The PICmicro family meets the specifications contained in the Microchip Data Sheet.
- Microchip believes that its family of PICmicro microcontrollers is one of the most secure products of its kind on the market today, when used in the intended manner and under normal conditions.
- There are dishonest and possibly illegal methods used to breach the code protection feature. All of these methods, to our knowledge, require using the PICmicro microcontroller in a manner outside the operating specifications contained in the data sheet. The person doing so may be engaged in theft of intellectual property.
- Microchip is willing to work with the customer who is concerned about the integrity of their code.
- Neither Microchip nor any other semiconductor manufacturer can guarantee the security of their code. Code protection does not mean that we are guaranteeing the product as "unbreakable".
- Code protection is constantly evolving. We at Microchip are committed to continuously improving the code protection features of our product.

If you have any further questions about this matter, please contact the local sales office nearest to you.

Information contained in this publication regarding device applications and the like is intended through suggestion only and may be superseded by updates. It is your responsibility to ensure that your application meets with your specifications. No representation or warranty is given and no liability is assumed by Microchip Technology Incorporated with respect to the accuracy or use of such information, or infringement of patents or other intellectual property rights arising from such use or otherwise. Use of Microchip's products as critical components in life support systems is not authorized except with express written approval by Microchip. No licenses are conveyed, implicitly or otherwise, under any intellectual property rights.

Trademarks

The Microchip name and logo, the Microchip logo, FilterLab, KEELOQ, microID, MPLAB, PIC, PICmicro, PICMASTER, PICSTART, PRO MATE, SEEVAL and The Embedded Control Solutions Company are registered trademarks of Microchip Technology Incorporated in the U.S.A. and other countries.

dsPIC, ECONOMONITOR, FanSense, FlexROM, fuzzyLAB, In-Circuit Serial Programming, ICSP, ICEPIC, microPort, Migratable Memory, MPASM, MPLIB, MPLINK, MPSIM, MXDEV, PICC, PICDEM, PICDEM.net, rfPIC, Select Mode and Total Endurance are trademarks of Microchip Technology Incorporated in the U.S.A.

Serialized Quick Turn Programming (SQTP) is a service mark of Microchip Technology Incorporated in the U.S.A.

All other trademarks mentioned herein are property of their respective companies.

© 2002, Microchip Technology Incorporated, Printed in the U.S.A., All Rights Reserved.

Printed on recycled paper.

DNV Certification, Inc.
USA
DNV MSC
The Netherlands
Accredited by the RvA
ANSI-RAB QMS ACCREDITED
DNV
ISO 9001 / QS-9000
REGISTERED FIRM

Microchip received QS-9000 quality system certification for its worldwide headquarters, design and wafer fabrication facilities in Chandler and Tempe, Arizona in July 1999. The Company's quality system processes and procedures are QS-9000 compliant for its PICmicro® 8-bit MCUs, KEELOQ® code hopping devices, Serial EEPROMs and microperipheral products. In addition, Microchip's quality system for the design and manufacture of development systems is ISO 9001 certified.